Real Audio For AM

By Bill Kleronomos, KD0HG
Rocky Mountain AMI Director

As many AMers move into the real radio world of using converted broadcasting equipment, I thought a quick hands-on primer on the ins-n-outs of broadcast audio might be in order within the friendly confines of the pages of ER. By no means can this be a fully comprehensive article because audio processing is a lot like making spaghetti sauce or chili. Everyone has their own recipe and tastes. I've found there is no such thing as a plug-n-play approach; if there were, every radio station on the dial would have stunning fidelity and hair-blowing loudness. Certainly, that's not the case, even with a "professional" radio budget.

THE GOES INTAS AND THE GOES OUTAS

The audio path into traditional ham gear has always been different from that that used in the professional audio field and uses completely different standards. Amateur and consumer audio gear is set up with a single-ended, unbalanced audio path that runs in the few millivolt range, up to maybe -10 dbm.

As a point of reference, 0 dbm refers to the voltage required to develop 1 milliwatt across a 600-ohm load. That calculates out to around .775 volts, RMS, so a –10 dbm consumer level would be close to a quarter of a volt, RMS.

Audio levels used with professional audio gear are considerably higher, offering much better noise immunity. Most gear these days is made to operate with either the +4 dbm or +8 dbm pro audio standard. This is referred to as "Line Level". The other standard that's important to us here is "Mike Level", which runs around –40 dbm.

HOW MUCH?

Even in this 21st century, the best way to measure analog audio levels is still with an analog meter. The industry-standard Simpson 260 is still made, 1950s-1960s vintage Triplett, Heath, Knight, and Simpson VOMs are commonly available at flea markets, garage sales and hamfests. Almost all of them have AC meter scales directly calibrated in db across 600 ohms, which makes audio measurements a no-brainer. There's a lot of surplus HP test equipment out there for those with deeper pockets.

One other essential for setting up audio processing and AM transmitters is a good audio generator, and owning an HP distortion analyzer like the model 330A is a definite plus. In addition to this simple gear, many of the better-equipped broadcast facilities have the audio generator and analyzer set made by Potomac Instruments. The Potomac AA-51 series of meters will do everything but butter your toast; in addition to making absolute level measurements, you can make individual measurements of THD, IMD, S/N, analog tape wow and flutter, all on a fast responding auto-nulling meter. If you can find one at an affordable price, grab it!

TYING IT TOGETHER

By definition, a balanced, 600-ohm pro audio system uses balanced cabling and connectors. A number of companies manufacture single and multiple pair shielded cable, a quick look at the catalogs of parts houses will turn up plenty of examples. As opposed to the unbalanced audio systems we're familiar with from consumer electronics, the use of shielded cable is usually unnecessary for indoor wiring at line level; the common-mode noise immunity of modern pro audio gear is amazing.

You can walk into almost any radio station and find interconnections done with nothing fancier that standard twisted pair telephone wire; and in many cases the same can be done in the home ham shack. You can even use 18-gauge zip cord from the local hardware store!

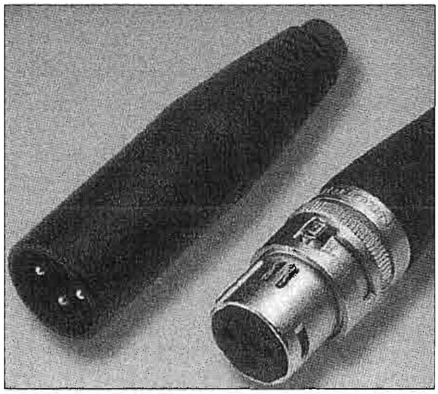
Yes, there are fancier cables than zip cord used with mike level audio levels and that's the only place where the use of a shielded cable is mandatory. Probably the ultimate mike cable is called "Star-Quad" and made by a company called Canare. It's not cheap at almost fifty cents a foot, but it's amazing stuff, using four conductors in a double-balanced configuration within a real braided shield. It's highly durable, flexible, kink-resistant and comes is a rainbow of available colors, useful in studio situations where a bundle of wires might snake around a facility and fast identification is necessary. Belden also makes a similar cable under their part #1192A. This double-balanced cable is so good that you can run it right next to the whip antenna of a 30-watt VHF or UHF "Marti" transmitter used for remote broadcasts and have no RF pickup at all into sensitive microphone preamp circuitry.

Using low-Z balanced audio wiring has another benefit- the ability to use extremely long microphone and line-level audio cables before losses. Noise and high frequency roll off then become a problem. It's not uncommon to use microphone cable that are several hundred feet long, for example, and still have virtually perfect broadcast audio. Try *that* with a D-104!

CONNECTORS

There's two common connectors in use with pro audio gear- the three-pin XLR style and the three-conductor stereo, tip-ring-sleeve style of ¼" plug. It couldn't be easier.

Compare that to the multiple, confusing and often hard-to-find styles used with consumer electronics and traditional ham gear. Japanese radios use an assortment of their own four, five and more pin connectors, American gear, especially our beloved boat anchors have used at least a dozen different styles and types of audio connectors and standards that I can think of, a number of which are getting pretty tough to find.



A typical 3-pin XLR connector as described in the text

There are very simple wiring standards used with pro audio:

If one is using an XLR connector, the balanced pair connects to pins #2 and #3; pin #1 is always cable shield and chassis ground. If you're using the ¼" style of connector instead, the balanced pair connects to tip and ring, the cable shield connects to the sleeve. One more thing, the pins on an XLR cable always "point" towards the direction of audio signal flow; the **output** jack of a mic or a piece of electronics is always a male connector, the **inputs** to a piece of gear are always female connectors, so under normal circumstances, the only XLR cables ever required around the station would be those with a male on one end and a female on the other. As I said, it couldn't be easier!

MICROPHONES FOR AMATEUR AM USE

This is as subject as open to opinion as anything out there, but I'll try to address general questions and confine the dis-

cussion to popular and economical broadcast-style microphones that most of us can afford to use.

Over the years many microphone designs have been used in broadcasting, from carbon microphones in the earliest days, to the superb sounding ribbon mics popular in the mid-20th century, to the modern dynamic and condenser mics. While popular with amateurs, there are several microphone designs that never made it into broadcasting- these include the crystal and related ceramic styles and the more modern inexpensive and compact electret element most commonly used in the two-way radio field; FM walkie-talkies, telephones and answering machines.

The differences between the different designs can be pretty much summed up by our desire for the accurate transformation of sound waves carried in air into electrical currents. Regardless of type or brand, the use of a unidirectional cardioid pattern is almost essential for managing background noise, room echo and acoustic feedback when the audio is run through broadcast-style high-gain compression or other processing. Trying to process the output of an omni directional microphone and have it sound decent can be a true exercise in futility!

Let's take a look at some of the pros, cons and useful characteristics of the different flavors of today's pro and semi-pro microphones.

CONDENSER MICROPHONES

The condenser microphone is probably the ne plus ultra of modern microphone design. Used by both broadcasters and in the recording studio, condenser mics can range from near $100 to many, many thousands of dollars. Modern condenser designs possess a number of valuable virtues: Extremely low distortion and a tremendous sensitivity to audio detail and the subtle nuances of voice and instruments. They are renown for their uniform frequency response, low distortion and clarity when reproducing transient sounds. Condensers are the first

The famous Neumann U87 condenser microphone which is noted for it's nearly perfect reproduction characteristics.

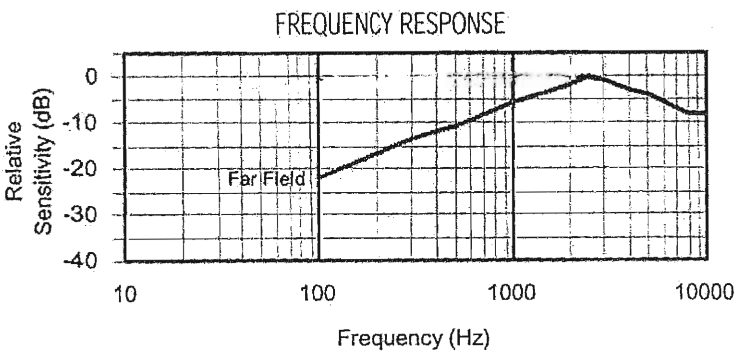
choice in many studios for recording female vocals and acoustic instruments- one not only hears the sound of a plucked guitar string, but you can clearly hear the musician's fingers manipulating the string and the pick plucking it. In my opinion, the most "real" sounding microphones out there.

In general terms, condenser microphones are made using a lightweight membrane and a fixed plate that act as

The Shure KSM-27 which is sold as an affordable studio cardioid condenser microphone

Another good looking affordable condenser mic is the Audio-Technica

opposite sides of a capacitor. Sound pressure against this thin metalized polymer film causes it to move. This movement changes the capacitance of the circuit, creating a changing electrical output. Most true condenser microphones need a 48-volt DC source of power to operate; this voltage is referred to as "phantom power". It is generally fed to the microphone through its output cable with the use of decoupling networks. You cannot use most true condenser microphones *without* a preamplifier that can provide a source of phantom power, so check first before considering one. One other thing to remember about condenser mics is that they tend to sound the same regardless of how closely you talk into them, having minimal "proximity effect". For



Above, the frequency response of a high quality electret hand microphone element. The facing page has the curves for the EV RE-1000 cardioid condenser mic.

The Shure KSM-44 is an example of a top-of-the line studio condenser mic featuring multiple pattern selection.

many people, the effect of listening to your own voice on a condenser mic can be almost un-nerving; there can be almost too much detail!

Perhaps the most revered studio condenser microphone out there is the Neumann U87. Manufactured since the mid-1960s, their current retail price runs around $2,500.

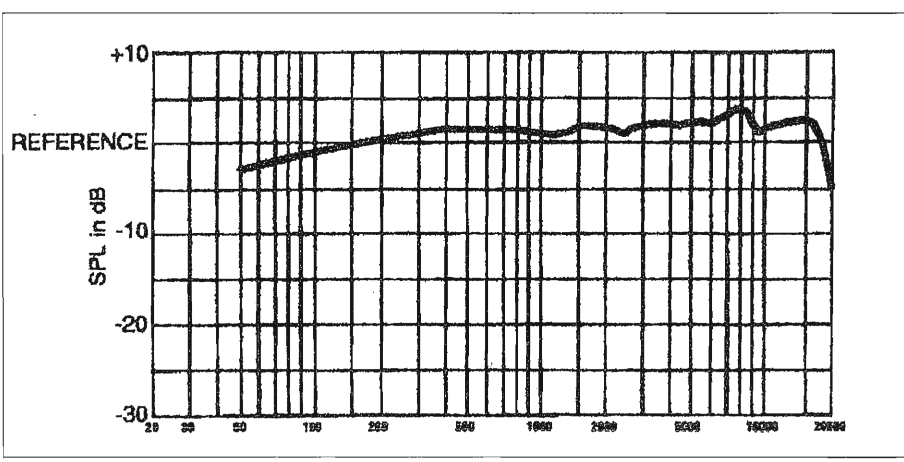
Studios occasionally send them out for aftermarket modifications, which can add several thousand dollars more to the price. Historically, condenser microphones have been completely out of the price range of the hobbyist, but there are now several lower-priced models on the market that are considerably more affordable.

While both Germany and Japan have traditionally been big players in the manufacture of condenser microphones, the Chinese have recently become a major manufacturing source and have brought the price of decent condenser mikes down into the affordable realm.

There is the Marshall MXL 603 condenser at $155, and the Rode NT-1 at $199, among others. Both have desired cardioid patterns.

THE ELECTRET MICROPHONE

The electret is nothing more than a simplified condenser mike, the principle difference it that the plates of the mike element carry a permanent electrical charge, much as a magnet carries a permanent magnetic "charge". Because of this, no external polarizing voltage is required to make the element work, but like its condenser mike cousin, the output is low and a low-noise, high-Z preamp is required to make one play. This preamp is frequently built into the mike cartridge itself.



Here is the reasonably priced, classic Electro-voice RE-1000 supercardioid condenser microphone.

The principal advantages to the electret element are low cost, small size and being able to be easily mass-produced. They are typically used is consumer applications ranging from telephone answering machines to FM walkie-talkies where their compactness and low cost are useful virtues. Historically, their main disadvantages have been a 'peaky' audio response, rather high distortion and substantial roll off at lower audio frequencies. A diaphragm the diameter of a pencil eraser simply won't have much bottom-end response! Electret elements are generally classified as omni directional elements, but they can be used in an enclosure that imparts some directionality.

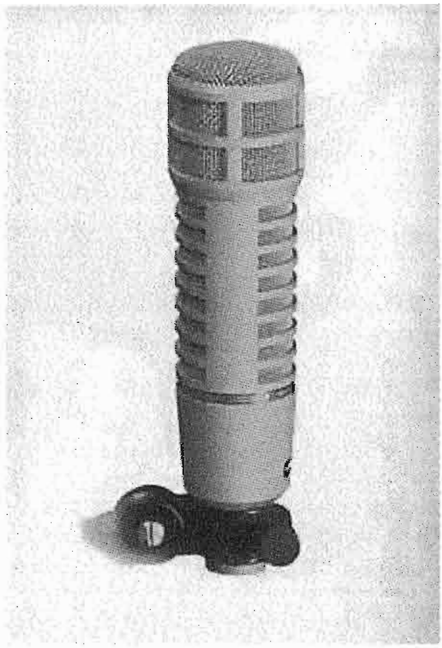
THE DYNAMIC MICROPHONE

Dynamics are the workhorse of the broadcasting world. Manufactured in an astounding variety of designs, a dynamic mike can be specifically designed for almost every application.

Dynamic mikes are nothing more than small electric generators, a coil of wire

The Sennheiser 421 dynamic cardioid microphone

attached to a diaphragm moves through the field of a small, powerful magnet generating a small electric current. The output of most dynamics is inherently low-impedance, ranging from as low as 50 ohms through the low hundreds of ohms. This property is useful as their low source impedance permits them to



Here's an affordable, quality dynamic cardioid, the Electro-voice RE-20

be used with very long lengths of balanced mike cable without significant sonic degradation. In broadcasting, microphone cord lengths of up to several hundred feet are occasionally used.

In addition to being highly durable and shockproof, dynamic mikes are made in an incredible variety of patterns and audio responses.

Some of the better ones, such as my favorite Sennheiser 421 have a switch built in to adjust their frequency response.

The directionality of dynamic mikes is largely determined by the design of the mike housing. Schemes such as ports and vents are used along the sides of the element, carefully tuned so that the pattern remains the same over the entire audio spectrum. In general, the longer the housing, the more directional the mike can be made. An extreme example of this is the "shotgun" mikes used to broadcast sporting events or to pick up specific sounds a long distance away.

There are other factors involved as well. For example, most of the Shure dynamic mikes such as the SM-7 and SM-57 have a great deal of proximity effect. Talking closer, almost 'kissing' the element greatly accentuates the bass response. A good radio DJ will take advantage of this and 'work' around the mike. Many of us have seen film of the best DJs working in their studios. Watch guys like Don Ingram, Larry Lujack, and the legendary Wolfman Jack and even today's Howard Stern. They'll scream, rant and dance around that mike, but when they want to make a point they'll come up close and talk softer, bringing up that bottom end and sounding like God in a 1960s Charlton Heston sandals-and-swords movie.

On the other hand, dynamics like the Electro-Voice model RE-20 have very little proximity effect by comparison; they sound pretty much the same regardless of operator distance. This makes them a very good choice for situations where several people might be sitting around a table during a radio talk show or newscast.

You can find a huge variety of dynamic mikes in almost every catalog that deals in broadcast or musician's supplies, and in a limitless price range. One overlooked source of inexpensive mikes is often the local music store or musician's hangout. Garage bands are constantly breaking up and their equipment often ends up for sale a bargain prices. Check out the musician's bulletin boards in these stores for the occasional bargain!

Well, that pretty much covers the front end of a quality AM station. The next installment of this article will cover mike preamps and audio processing. *Stay Tuned!*

[Editor's note: Bill will present part two of this article in a future issure of ER.]

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Real Audio for AM, Part 2

Bill Kleronomos, KDØHG
heavymetalrally@earthlink.net

The last installment of this article covered the front end of a high fidelity, low distortion AM station— your microphone. Remembering the old adage, "Garbage in = Garbage out", it's very difficult to overcome the handicap of using a deficient mike no matter what sort of subsequent processing in the audio chain might be used.

Audio processing is an art form. There is no cookbook that'll tell you the precise types of equipment to use and the correct settings. Much like making chili, it's much a matter of individual taste and what ingredients are available. It's good to bear in mind one thing: The technical definition of distortion is that the output of an electronic device is not a replica of its input. Audio processing is deliberately introducing distortion on the signal; the art is to make that distortion sound both pleasing and intelligible.

You should begin with an idea of what your goals are. Do you want an open, "pin drop" sort of quality, or a blasting wall of audio? Do you want an intense audio effect, or do you want to sound laid back? Are you willing to accept more perceived distortion in order to squeeze out the last drop of loudness? One good way to start is to identify those stations that have sonic qualities that you like and then try to sound better.

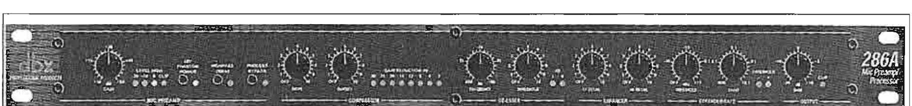
"Standard" equipment line-up

There is a specific order, or path, which is best followed in the audio chain feeding a transmitter. The order generally is:

- Microphone Preamp
- Compression
- EQ
- Clipping and filtering

The simple function of a preamp is to amplify the –30 to –60 dbm output of a typical broadcast mike to line levels of 0 dbm to +8 dbm. There are many preamps available on today's market at excellent prices. The usual attributes apply, most important are low noise and distortion. A simple balanced-input op-amp circuit can be homebrewed to provide the required gain of around 50 db. Lately, vacuum tubes have made a big comeback in mike preamps. It's hard to beat the sound of a 12AX7!

With their differential inputs and excellent common-mode noise rejection, op-amps are an excellent choice to use as preamps. In the past, their principle drawbacks have been high levels of hiss and noise as well as restricted frequency response in higher gain circuits. Don't use a 25-year-old 741! Much better choices are the more modern FET-input devices like the Texas Instruments TL-072. In my opinion, the Burr-Brown (B-B) series of op-amps are the best sounding devices out there. Their internal FET amplifiers mimic the transfer characteristics of vacuum tubes and they have distortion figures that were a dream not so long ago. I highly recommend the B-B OPA134/2134/4134 single/dual/quad amplifiers. They can also be used as a drop-in upgrade for older audio gear using lesser devices. They're available at



The DBX 286A mic preamp and processor, as discussed in the text, is a very capable and affordable unit with many advanced features.

suppliers like Digi-Key for around two bucks.

There are a few bargain mike preamps and processors out there, too. The DBX model 286-A is one. I think it's the Swiss army knife of economy microphone processors. This one rack-space high unit includes a preamp with switchable phantom-power for condenser mics, a compressor, high and low frequency EQ, expander and gating (essential for knocking back gain between words and reducing background noise). It also includes an insert jack for adding other audio accessories (like a reverb!). This wonder box retails for $299, but can be purchased at discount suppliers for prices as low as $185. I've seen them for less than $100 on Ebay as well. As I previously mentioned, your local musician's store can be your best friend when scrounging for this sort of pro audio gear. [Editor's note: 1 rack space = 1 ¾ inches in height]

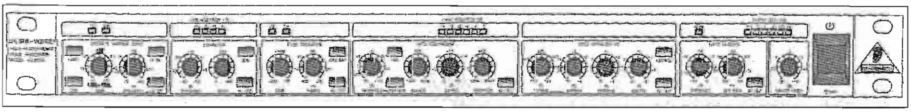
Compression

Audio compression is probably best defined as a form of automatic gain control with the goal of maintaining constant speech intensity over varying mike audio levels. That should be the main goal of any compressor stage. Speech compression has long been touted as a means of increasing punch and intelligibility, but the reality is that most compression schemes can offer only a db or two of perceived improvement in audio punch. Simple compressors used by hams usually have attack times of around 10 milliseconds and decay times of 300 milliseconds or longer. At 1 KHz, the middle of the voice spectrum, a cycle is only 1 millisecond; so traditional compressors are not fast enough to practically limit peaks.

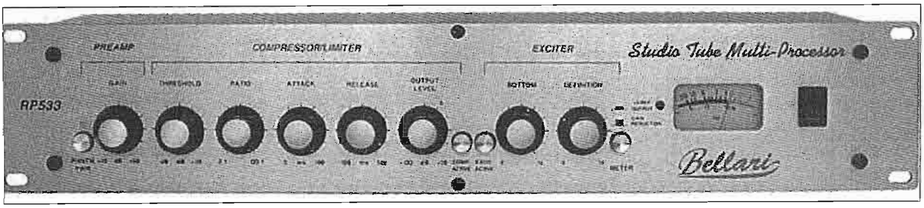
For our purposes, the dynamic range of the human voice is about 35 db but room acoustics and background noise limit the practical amount of compression we can use to 15 db or so. A very useful feature many modern compressors have what is called "gating". A simple compressor will run its gain up to the maximum between words and silent periods in normal speech, raising background noise and causing an unnatural 'breathing' effect.

Who hasn't heard those sideband stations that have so much room and blower noise in the background that their intelligibility is impaired? That's solely due to the simple compression action of the ALC incorporated into SSB equipment. By comparison, a *gated* compressor will recognize that audio inputs below a certain threshold are not to be boosted so those moments of silence pass through the system with the system gain reduced. The gate acts almost like the inverse of a compressor; its logic is, "If there's no audio, turn the gain *down*!"

As I mentioned, traditional compressors act to automatically "ride the gain" on speech, offering minimal improvements in loudness. By the use of a fully adjustable compressor and extremely fast attack times in the microsecond range, a compressor begins to cross a functional dividing line and begins to act as a speech clipper. Many people have instinctively presumed a direct relationship between loudness-punch and the amount of compression used. This is not as much of as factor as is the relationship between loud-



The Berringer VX2000 Ultra-Voice is another affordable mic processor with high performance. Among other features is an input "tube emulation" stage and discrete devices in the preamp for low distortion.



The Bellari RP533 is not inexpensive, but it is an advanced vacuum-tube based design that features a fully adjustable compressor/limiter with calibrated attack times from .5 to 100 ms. As with most of the professional models the release time is adjustable, as is the compressor level.

ness and a compressor's time constants. The faster your compressor acts, the more loudness it can provide. *To get louder, you don't turn up the gain on a compressor; you speed up its attack time.*

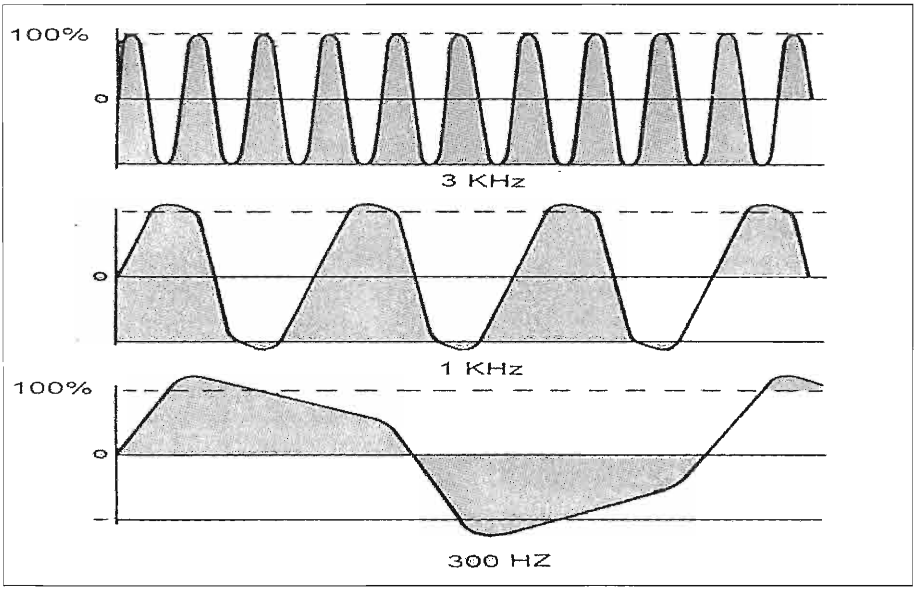
EQ

Equalization is an important factor in how a station sounds. Having a generally flat response, broadcast microphones can sound muddy when used without EQ tailored to the desired response. A bit of upper mid-range boost in the 1.5 to 2 KHz frequency band can help improve intelligibility and punch, but too much can give a station what I call "AM honk", or a nasal, 'cheap speaker' sort of sound. It's a dicey proposition to use too much EQ without following it with a speech clipper or peak limiter stage. With 6 db of mid-range pre-emphasis, for example, the mid-range will hit the audio stages and ALC on an SSB transmitter before the rest of the audio spectrum can, however, it is the lower frequencies that carry the bulk of the energy in speech and move the meters more. By the time someone turns up the mike gain so that there is a healthy amount of ALC or compression showing on the meter, there's a danger that the emphasized mid frequency components will have already gone into clipping and distortion because of their greater amplitude. This is something to be careful of when either dialing in EQ or using a microphone with any large degree of mid or upper range emphasis. You might gain punch and perceived loudness, but at the expense of generating clipping and distortion products in the transmitter's audio and RF stages.

Once again, there are always tradeoffs between loudness and distortion, and this one example how a "punchy" microphone or EQ can often end up sounding rather ratty.

CLIPPING, LIMITING and FILTERING

The final stage in any audio processing scheme should be a clipper-limiter, followed by high pass audio filtering. A clipper can be nothing fancier than a pair of diodes with adjustable biasing, and there's plenty of used equipment on the market at decent prices, such as the famous CBS Volumax series. Clipping is realistically the best way to improve a station's audio loudness- remove the audio signal's peaks and build up in a relative manner the effective level of the weaker sounds. Many consonant sounds such as s, t, k, b, v and l in human speech are up to 30 db down from the vowels, yet speech would severely lack intelligibility without those sounds being well articulated. Numerous tests have indicated that 6 db of audio clipping begins to become perceptible, 12 db of clipping adds significant loudness and levels up to 25 db are considered tolerable on speech. For that matter, it has been demonstrated that speech clipped 100%, without having any amplitude variations



The effect of phase shift and insufficient modulator low-frequency response which causes overmodulation and splatter is called "low frequency twist". These curves show how differing components passing through the same modulator are progressively distorted as their frequency is lowered.

at all is entirely intelligible. Clipping around 15 db doesn't sound *quite* natural, but it is not unpleasant to listen to. That's where I typically run the clipping level on my AM ham station.

There are a number of pitfalls to be aware of in the use of a clipper. First, clipping, by definition, is distorting the audio waveform. Distortion is the addition of harmonic energy to the fundamental. Clipping converts sine waves into modified square waves, and a square wave consists of the fundamental along with its odd-order harmonics- 3rd, 5th, 7th, 9th etc. Not only are these odd-order harmonics unpleasant to hear, but they also quickly exceed our desired frequency bandwidth without adding anything worthwhile. While you can filter everything above 3 KHz, for example, what about the distortion products generated by clipping say, a 150 Hz component? That crud will fall at 450, 750, 1050,1350 Hz and up, right in the middle of the speech audio spectrum. **That** certainly won't improve intelligibility!

This is one of the reasons why pioneers like Mike, KO6NM, developed multiband audio processing many years ago. The idea is to split audio into several bands, independently process and filter each band, then re-combine the processed and filtered audio. Even when found used, multiband processors like the Orbans are expensive, but they're the only way to achieve the ultimate in loudness, articulation and intelligibility by permitting higher levels of processing than could otherwise be effectively used.

In any case, regardless of the type of setup, a low-pass filter *must* be used after the final clipper-limiter or otherwise you're guaranteed to be broad as a barn on our bands. Hi-Fi AM is one thing, wastefully transmitting unintelligible distortion products across 15 KHZ or

more of the 75-meter band is another!

To be completely effective, the use of a clipper-filter before the audio driver stages of an AM transmitter requires that the phase shifts in the transmitter's modulator be kept to a minimum as the waveform gets "tilted" and can cause over modulation and splatter. While the real solution is to improve the LF response of the transmitter's audio stages, rolling off the bass response of the microphone by the use of the EQ to better match the bass response of the transmitter's modulator can minimize the problem.

The final clipper-limiter stage is where techniques such as asymmetrical modulation can be applied, where positive modulation peaks are accentuated to the limits of the modulator's power while the negative, splatter-causing peaks are held to –100% maximum. The CBS Volumax 4000, for example, has a front panel switch that adjusts the amount of asymmetry in the audio fed to the transmitter modulator. Many modern AM broadcast stations run positive peaks of around 125% to improve their signal-to-noise ratios. Experience has shown that even with unlimited modulator power available, the practical limit for asymmetry is around 125% on speech. Modulation percentages beyond that point can sound distorted on an AM envelope detector as there is insufficient carrier for proper detection. Signals sound marginally distorted like an SSB signal would with insufficient BFO injection on an older receiver. I have experimentally tried positive modulation percentages of up to 150% on a modern digitally-modulated AM broadcast transmitter and found the apparent loudness to be impressive, but with equally impressive amounts of distortion in the detected audio. Once again, the loudness vs. quality issue rears its ugly head. There's just no avoiding it!

There are several other ways of achieving asymmetrical modulation such as the use of negative cycle loading and the "Ultramodulation" circuit in a Class B modulator. Those subjects were previously covered in my article "A Legal Limit Modulator" back in July 1989, ER #3. Many of the audio purists among us do not accentuate positive modulation peaks, and admittedly doing so may add a slight degree of distortion. My own approach is a pragmatic one. If interference or noise is an issue, forget about the Hi-Fi and let the audio rip! There are few things more annoying than trying to copy an AM signal with a decent carrier but "PW" audio!

A whole book could be written on AM audio processing and equipment; admittedly I've only scratched the surface in this article but I need to end it somewhere. If there is an aspect to this subject that might be of general interest for a future article, contact the editor and ask!

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